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Transmitted herewith for filing is the patent application of

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For : **Method and System for Customer Selected Direct Dialed Voice-Over-Internet Protocol (VOIP)**

Enclosed are:

1. **46** sheets of specification including **20** sheets of claims, and **1** sheet of abstract.
2. **7** sheet(s) of **drawings (FIGS. 1-7)**
3. Declaration and Power of Attorney
4. Assignment Cover w/ Assignment and Agreement

The filing fee has been calculated as shown below:

	NUMBER FILED		NUMBER EXTRA*	RATE ($)	FEE ($)
BASIC FEE				690.00	$ 690.00
TOTAL CLAIMS	70	- 20 =	50	18.00	900.00
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MULTIPLE DEPENDENT CLAIM PRESENT				260.00	0.00
FEE FOR RECORDATION OF ASSIGNMENT				40.00	$ 40.00
*Number extra must be zero or larger				TOTAL	$1,942.00
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5. The Commissioner is hereby authorized to charge the filing fee of **$1,942.00** to Kenyon & Kenyon Deposit Account No. **11-0600.** The Commissioner is further authorized to charge any additional fees or credit any overpayments to the above deposit account number. A copy of this letter is being submitted to facilitate processing of this application.
6. A. Any additional filing fees required under 37 C.F.R. § 1.16.
   B. Any patent application processing fees under 37 C.F.R. § 1.17.

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METHOD AND SYSTEM FOR CUSTOMER SELECTED DIRECT DIALED VOICE-OVER-INTERNET PROTOCOL (VOIP)

TECHNICAL FIELD

The present invention relates generally to Voice-over-Internet protocol (VOIP) systems, and more particularly to direct-dialed, that is, "single-stage" VOIP systems.

BACKGROUND OF THE INVENTION

Voice-over-Internet protocol (VOIP) telephony systems deal with the transmission of voice-band calls over a packet data network, such as a corporate intranet or the Internet. Using current VOIP systems to make long distance calls offers a number of interesting advantages when compared to traditional long distance telephone service. Some of these advantages include, for example, the ability to place long distance calls as two local voice band calls using an IP data link between them, one local call at the calling number end and one at the destination number end; and being able to manage a voice and data network as a single network. Likewise, additional advantages include moving, adding and changing Internet Protocol (IP) phones is easier and cheaper than regular telephones; providing new and integrated services

including integrated messaging, bandwidth on demand, voice e-mails, "voice portals" that provide interactive voice response access to systems such as the Internet; and simplified setting up, tearing down and transferring of phone calls.

Unfortunately, current VOIP systems have some significant disadvantages as well, including no single-stage dialing as on a circuit-switched Public Switched Telephone Network (PSTN); only two- or three-stage dialing capabilities; and no integrated billing system that can detect and track network use and associate the use with a number for billing purposes. "Single-stage" dialing, which is also known as direct-dialing, permits a caller to dial the desired destination number, the telephone network automatically recognizes the telephone number from which the caller initiated the call, the telephone network automatically determines which calling plan(s) the caller has been registered and routes the call based on the caller's calling plan(s).

"Two-stage" or "three-stage" dialing both require the caller to first call an IP network access number, which can be either a local or a national number, and either a toll-free or a toll number. Next, for two-stage dialing, the system at the access number either automatically detects and recognizes the caller's number or the caller's phone service is programmed to automatically send the caller's account number and PIN when the caller dials the access number; the system then connects the caller to the system and the caller enters the desired destination number. Alternatively, for three-stage dialing, after the system at the access number answers, the caller is usually prompted to enter the caller's account number and PIN and, then, the caller enters the desired destination number.

An example of both a two-stage and a three-stage dialing, prepaid VOIP calling plan is AT&T's Connect 'N Save® service. In this two-stage dialing service, the customer enrolls in the service, prepays a set amount using a

credit/debit card or a check, and signs up for the Express Login feature at the customer's registered telephone number. The Express Login feature only operates from the customer's telephone number and automatically identifies the customer's account number and PIN when the customer calls the access number from the customer's registered telephone number. After connecting to the service, the customer dials the desired destination number and waits for the call to connect with the destination number. The Connect 'N Save® service has both local and national access numbers that can be used to access the service. An example of three-stage dialing occurs in the Connect 'N Save® service if the customer either did not sign up for the Express Login feature or is calling an access number from a telephone number other than the one that was used to sign up for the Connect 'N Save® service. In this case, the customer first dials the access number; second, enters the customer's account number and PIN; and third, dials the desired destination number.

FIG. 1, depicts a block diagram implementation of current two-stage and three-stage voice-over-Internet protocol (VOIP) services. In FIG. 1, customer (calling party) telephones 115 and 116 are connected to a local access provider network 118, which is in turn connected to a circuit-switched, Public Switched Telephone Network (PSTN) 145 and to an IP gateway 120, which is communicatively linked to an IP Network 110. The IP gateway 120 provides both local and toll free number VOIP access to the IP Network 110, however, single telephones are shown accessing the IP gateway 120 for each of the local and toll free VOIP access numbers in FIG. 2 for reasons of clarity. The IP Network 110 is communicatively linked with another IP gateway 125, which provides VOIP service to U.S. locations, and the IP gateway 125 is communicatively linked to a destination telephone 135 through a destination local access provider network (not shown). Similarly, the IP Network 110 is

communicatively linked with another IP gateway 130, which provides VOIP service to international locations, and the IP gateway 130 is communicatively linked to a destination telephone 136 through another destination local access provider network (not shown). The IP Network 110 is communicatively linked with another IP gateway 140, which is communicatively linked to a second circuit-switched PSTN 145. When a calling party places a call to a destination telephone number that is not served by any of the VOIP service IP gateways 125 and 130, the call is routed through the IP gateway 140 to the second PSTN 145 for completion as a circuit-switched telephone call.

In FIG. 1, when a caller desires to place a VOIP call the caller first calls one of the access numbers at the IP gateway 120 from one of the telephones 115 and 116. These access numbers can be either a local or a national numbers, and either a toll-free or a toll number. Next, for two-stage dialing, the system at the IP gateway 120 either automatically detects and recognizes the caller's number or the caller's phone service can be programmed to automatically send the caller's account number and PIN when the caller dials the access number, connects the caller to the system and the caller enters the desired destination number. Alternatively, for three-stage dialing, after the system at the IP gateway 120 answers, the caller is usually prompted to enter the caller's account number and PIN or credit/calling/debit card number and PIN and, then, the caller enters the desired destination telephone number. Upon receiving the destination telephone number, the IP gateway 120 determines an appropriate destination IP gateway, for example, destination IP gateway 125, converts the call into packets and then routes the packets to the destination IP gateway 125 through the IP Network 110. The IP network 110 is configured to receive the packets and then route the packets to the destination IP gateway 125 through the destination local access provider network (not

shown). The destination IP gateway 125 receives the packets, reassembles the packets, converts the packets back to a voice-band call and sends the reassembled voice-band call to the destination telephone number 135 through the other destination local access provider network (not shown).

Unfortunately, the currently-available two-stage and three-stage VOIP services require the entry of multiple phone, account and PIN numbers to use the service, which is inefficient and unpleasant for customers. Another disadvantage of some currently-available two- and three-stage VOIP services is that they require either the purchase of new prepaid calling cards or the replenishment of minimum prepaid account balances because the calling party's regular telephone service billing system is not connected to and does not communicate with the VOIP service.

Accordingly, a single-stage VOIP system is desired that enables the user to directly dial a destination number from a telephone attached to the circuit-switched PSTN, where the call is automatically routed as a VOIP call over the IP network and billed to the calling party's regular telephone-bill account. Likewise, a provisioning system is desired that receives customer orders for the VOIP service, provisions the network and billing systems based on the orders and maintains the operational and informational synchronization between the network and billing systems.

SUMMARY OF THE INVENTION

The present invention is directed to a single-stage VOIP system that enables the user to directly dial a destination number from a telephone attached to the circuit-switched PSTN and have the call automatically routed as a VOIP call over the IP network and billed to the calling party's regular telephone-bill account.

In an embodiment of the present invention, a method for routing direct-dialed voice-band calls over an IP network includes receiving a direct-dialed voice-band call from a calling party telephone number, the direct-dialed voice-band call being associated with a destination number. The method further includes automatically routing the direct-dialed voice-band call to the destination number as a VOIP telephone call if the calling party telephone number is registered for a VOIP service and if the destination number of the direct-dialed telephone call is accessible by the VOIP service.

In an embodiment of the present invention, a method for automatically provisioning and maintaining a network system for routing direct-dialed voice-band calls from a calling party telephone number over an IP network includes receiving a VOIP service registration for the calling party telephone number, generating at least one order record for the calling party telephone number's VOIP service and storing the at least one order record for the calling party telephone number's VOIP service. The method further includes managing the billing interaction for a billed account between at least one calling party telephone number and a billed telephone number; synchronizing changes made to the stored at least one order record for the calling party telephone number's VOIP service, between the network system and a billing system, due to calling party activations, disconnections and changes; and processing at least one call detail record including at least a terminating access identification (ID).

In an embodiment of the present invention, an apparatus includes a first voice-band switch and a database coupled to the first voice-band switch. In the apparatus, the first voice-band switch is configured to receive a direct-dialed voice-band call from a calling party's telephone number and to automatically designate the direct-dialed voice-band call as a VOIP call.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG 1 is a block diagram implementation of known two-stage and three-stage voice-over-Internet protocol (VOIP) services.

FIG 2 is a block diagram of the direct-dialed VOIP system according to an embodiment to the present invention.

FIG 3 is a detailed block diagram of the direct-dialed VOIP system of FIG. 2 which illustrates a direct-dialed VOIP apparatus according to an embodiment to the present invention.

FIG 4 is a detailed functional flow diagram of the operation of the entire direct-dialed VOIP network system of FIG. 3, in accordance with an embodiment to the present invention.

FIG 5 is a detailed functional flow diagram of the operation detailed in FIG. 4 for the direct-dialed VOIP apparatus of the entire direct-dialed VOIP network system of FIG. 3 according to an embodiment to the present invention.

FIG 6 is a detailed block diagram of an offline provisioning system which is necessary for the proper operation of the direct-dialed VOIP service according to an embodiment to the present invention.

FIG 7 is a detailed functional flow diagram of the operation of a scalable offline provisioning system in FIG. 6 according to an embodiment to the present invention.

DETAILED DESCRIPTION

A direct-dialed voice-over-Internet protocol (VOIP) service is provided to registered customers so that the VOIP service is transparent to the customers when it is in operation. In accordance with an embodiment of the present invention, a customer registers (signs-up) for the VOIP service (calling plan)

and, once a customer registers for the service, then all of the customer's voice-band calls can be automatically routed through the VOIP service just as with other standard circuit-switched calling plans. Note that the registration of customers for the VOIP service is not required by embodiments of the present invention because the VOIP service could be offered as the standard service that customers would have to opt out of, that is, choose not to use, by registering for an alternate calling plan.

FIG. 2 is a block diagram of the direct-dialed VOIP system according to an embodiment to the present invention. In FIG. 2, a customer telephone 200 is connected to an originating local-access-provider network 205 and assigned a telephone number (not shown), which is stored at the originating local-access-provider network 205. More generally, the originating local-access-provider network 205 may provide local voice-band telecommunications services for numerous telephone numbers, however, only a single telephone 200, which is associated with a single telephone number, is illustrated in FIG. 2 for reasons of clarity. An alternate embodiment of the present invention is contemplated in which multiple telephones, each associated with a unique telephone number, are connected to and registered for the VOIP service. In another embodiment of the present invention, multiple telephones can be associated with the single telephone number registered for the VOIP service, for example, the multiple telephones can be extension phones that are all proximally located so that they can be connected to a single telephone line, which provides service for the single telephone number. The originating local-access-provider network 205 may be communicatively linked with a circuit-switched, PSTN 210 when a calling party places a call to one of destination telephones 135, 136. The destination telephones can be located both within and outside the service area of the originating local-access-provider network

205. For example, calls outside the local service area can include an intra-state long distance call, an out-of-state long distance call or an international long distance call.

The PSTN 210 is capable of providing service for numerous different calling plans and millions of different customers. In accordance with embodiments of the present invention, the PSTN 210 generally includes a long distance telephone provider network, a local telephone service provider and a Private Branch eXchange (PBX). PSTN 210 is communicatively linked with an IP gateway 215, and the IP gateway 215 is communicatively linked with the IP network 110. The PSTN 210 also is communicatively linked with other US and foreign destinations that are not served by IP gateways. The IP network 110 is communicatively linked with an IP gateway 125 for US locations served by this IP gateway, and the IP network 110 is also is communicatively linked with an IP gateway 130 for foreign locations served by this IP gateway. Note that while FIG. 2 and the above detail only show and describe single components for the sake of clarity, each component can be representative of multiple components. For example, multiple telephones (for example, telephone 200) each can be communicatively linked to the local-access-provider network 205. Likewise, multiple originating local-access-provider networks 205 can each be communicatively linked to the PSTN 210 of FIG. 2, or each of a plurality of originating local-access-provider networks 205 can each be communicatively linked to a separate PSTN 210. Similar multiple configurations of the IP gateways 215, 125 and 130, the IP network 110 and destination telephones 135 and 136 are also contemplated in embodiments of the present invention.

In FIG. 2, in accordance with an embodiment of the present invention, a caller places a direct-dialed long-distance call from the telephone 200, that is, for example, the caller dials 1-555-123-4567. The call is received by the

originating local-access-provider network 205 and then routed to the PSTN 210, because the "1-" dialing prefix identifies the call as a long-distance call. After receiving the direct-dialed long-distance call, the PSTN 210 determines if the call (or a portion thereof) can be routed as a VOIP call over the IP network 110. The call can be routed as a VOIP call over the IP network 110 if the caller's number is registered for the VOIP service and the destination number is served by the IP network. If the call can be routed as a VOIP call, the PSTN 210 initiates a VOIP billing record to track the duration and cost of the VOIP call and, then, the PSTN 210 routes the call to an IP gateway 215. Once the IP gateway 215 has received the call, the IP gateway 215 determines a specific destination IP gateway 125 or 130 based on the destination number, for example, IP gateway 125; converts the voice-band call setup information to packets, and then routes the packets to the destination IP gateway 125 through the IP network 110. IP network 110 is configured to receive the packets and then route the packets to the destination IP gateway 125. Destination IP gateway 125 receives the packets, reassembles the packets, converts the packets back to a voice-band call and sends the reassembled voice-band call to the destination number 135 through a destination local access network provider (not shown).

In an alternate embodiment of the present invention, an originating local-access-provider network can be configured to communicate directly with an IP gateway and perform the same functions as described above for the PSTN 210. Similarly, in another embodiment, an originating local-access-provider network 205 can be configured to communicate directly with an IP gateway 215 for calls within the service area of the originating local-access-provider network 205 (communication link 206).

In another embodiment of the present invention, an originating local-access-provider network can be a corporate telephone network or Private Branch eXchange (PBX) which is configured to communicate directly with an IP gateway and an IP network, which is a local area network (LAN).

FIG. 3 depicts a detailed block diagram of the direct-dialed VOIP system of FIG. 2 which illustrates a direct-dialed VOIP apparatus according to an embodiment to the present invention. In FIG. 3, elements that are common with FIG. 1 and FIG. 2 maintain their prior numbering schemes. In FIG. 3, telephone 200 is communicatively linked to the originating local-access-provider network 205, which in turn is communicatively linked to PSTN 210. In FIG. 3, in accordance with an embodiment of the present invention, PSTN 210 is shown to include a voice-band telephone switch 305, such as, for example, an Electronic Switching System Number 4 (4ESS) Originating Assist Switch (OAS), which provides the circuit-switched communication link with the originating local-access-provider network 205. The PSTN 210 also is communicatively linked to a database 310. Database 310 may be, for example which is implemented in FIG. 3 as a Universal Subscriber Data Structure (USDS), containing, but not limited to, customer telephone number information, such as, registered network service(s) for the telephone number, a country code listing of available destination IP gateways, and routing instructions. The OAS 305 receives an incoming call from telephone 200 and uses information associated with the incoming call and USDS 310 information to determine if the incoming call is to be routed as a VOIP or a circuit-switched call.

The OAS 305 also may be communicatively linked to a second telephone switch 307, such as, for example, a 4ESS Handoff Assist Switch (HAS), which is used to route the calls to be routed as VOIP calls, and a third telephone switch 315, such as, for example, a 4ESS International Switching

Center (ISC), which is used to route the calls to be routed as circuit-switched calls. In accordance with an embodiment of the present invention, the HAS 307 is communicatively linked to the originating IP gateway 215, which is, in turn, linked to an IP command center database 325, which contains a listing of the specific IP gateways or complex of gateways that serve the destination number. The IP gateway 215 uses the specific IP gateway information from the IP command center database 325 to determine the final routing instruction for the call, converts the voice-band call and setup information to packets and then routes the packets to the IP network 110. In addition to the routing function, the HAS 307 may create a billing record that is used to track and record the length of the VOIP call to billed to the caller's standard telephone-bill account. The HAS 307 was used, in this embodiment of the present invention, to consolidate and reduce the number of IP gateways 215 and the number of connections that are actually made to the IP gateway 215 since the IP gateway 215 can only handle a limited number of connections. For example, in the experimental design, the IP gateway 215 could only receive a single T-1 (Trunk Level 1) connection. In accordance with an embodiment of the present invention, the system is configured with 6 IP gateways 215 and each IP gateway 215 can receive four (4) separate T-1 lines.

In an alternate embodiment of the present invention, an HAS is not used and an OAS is directly and communicatively linked to an IP gateway and the OAS also is modified to perform the functions of the HAS described above. However, in this embodiment of the present invention, the OAS can only be located a finite distance (for example, approximately 900 miles) from the IP gateway.

The ISC 315 is communicatively linked to other circuit switches, at destinations that are not served by IP gateways, to route standard circuit-switched calls.

The IP network 110 routes the packets to the appropriate destination IP gateway 125, 130 where the packets are received, reassembled and then converted back to a voice-band call. IP gateway 125 is communicatively linked to an Electronic Switching System Number 5 (5ESS) 330 and IP gateway 125 routes the reassembled voice-band call to the 5ESS 330. The 5ESS 330 routes the reassembled voice-band call to a destination local-access-provider network 335, which, in turn, routes the reassembled voice-band call to the destination telephone number 135. IP gateway 130 is directly and communicatively linked to another destination local-access-provider network 340 and IP gateway 130 routes the reassembled voice-band call to the destination local-access-provider network 340. The other destination local-access-provider network routes the reassembled voice-band call to the destination telephone number 136.

In accordance with an embodiment of the present invention, an emergency backup network is associated with the direct-dialed VOIP service and is configured to operate when, the destination IP gateway 125 or 130 goes down, that is, becomes unavailable, after the direct-dialed call has been routed to the originating IP gateway 215. In this embodiment of the present invention, if, after the direct-dialed call has been routed to the originating IP gateway 215, the destination IP gateway 125 or 130 goes down, the service automatically routes the direct-dialed call to another IP gateway in the IP network 110 and then uses the Connect 'N Save® service to complete the call. The other IP gateway can be located anywhere in the IP network 110. In an embodiment of the present, preference is first given to the other IP gateway that is located the

nearest to the destination IP gateway 125 or 130. If the nearest IP gateway is also unavailable, then the call is routed through an IP gateway, which serves as the main IP gateway for the Connect 'N Save® service, to a 4ESS switch that, then, forwards the call to the ESS ISC 315 for completion as a circuit-switched call.

In FIG. 3, when a called party at a destination telephone number transmits a voice-band response back to the calling party, the above process is reversed and the transmitted voice-band response is routed through the local-access-provider network 335 or 340 and then to the appropriate IP gateway 125 or 130 where the voice-band response is converted to packets and routed over the IP network 110 to the originating IP gateway 215. When the IP gateway 215 receives all of the packets from the voice-band response, the IP gateway 215 reassembles the packets in the correct order; converts the packets back to a voice-band response and, then, routes the reassembled voice-band response to the PSTN 210 for routing through the local-access-provider network 205 to the calling party at telephone number 200.

FIG. 4 depicts a detailed functional flow diagram of the operation of the entire direct-dialed VOIP network system of FIG. 3, in accordance with an embodiment to the present invention. In FIG. 4, in block 405 a direct-dialed voice-band call is received by the OAS 305 from the calling party's telephone number, where the direct-dialed voice-band call is associated with a destination telephone number. In block 410, a test is performed to determine if the calling party's telephone number was registered for the VOIP service. In an embodiment of the present invention, before the test can be performed, the OAS 305 sends the calling party's telephone number and the associated destination telephone number to the USDS 310. After receiving these numbers, the USDS 310 does a lookup in a database using the calling party's

telephone number to determine if the calling party's telephone number has been registered for the VOIP service. If the calling party's telephone number was not registered for the VOIP service, then, the USDS 310 sets a terminating address value to indicate whichever service the calling party is registered, for example, "PSTN", or to a system determined default value if the calling party was not registered for any network services. The USDS 310, then returns the terminating address to the OAS 305 where the OAS 305 determines that the terminating address value indicates the voice-band call is to be routed as a circuit-switched network call and, then, in block 412, the OAS 305 routes the direct-dialed voice-band call to the ESS ISC 315 to continue routing the direct-dialed voice-band call as a circuit-switched call through the PSTN. In block 414, a standard circuit-switched billing record is initiated. If the calling party's telephone number was registered for the VOIP service, then, in block 415, a test is performed to determine if the destination number of the voice-band call is served by an IP gateway. If the destination number of the voice-band call is not served by an IP gateway, then, in block 412, the voice-band call is routed as a circuit-switched network call and, in block 414, a standard circuit-switched billing record is initiated.

If the calling party's telephone number was registered for the VOIP service, then, the USDS 310 then checks the destination telephone number against an allowable number to determine if the destination telephone is served by an IP gateway. If the destination telephone number is not served by an IP gateway, the USDS 310 sets the terminating address to indicate a non-VOIP service, for example, "PSTN", and then returns to the OAS 305 the terminating address. The OAS 305, then, determines that the terminating address value indicates the voice-band call is to be routed as a circuit-switched network call and, then, routes the voice-band call as described above for block 412. If the

destination telephone number is served by an IP gateway, then, the USDS 310 returns to the OAS 305 the terminating address value called an adjunct logical address (ALA) indicating a partial routing instruction to reach the VOIP network. In block 420, the OAS 305 receives the ALA partial routing instruction. Then, in block 425, the OAS 305 routes the voice-band call and partial routing information to the HAS 307. In block 430, the HAS 307 receives the direct-dialed voice-band call and partial routing information. The HAS 307 determines through which circuits to send the direct-dialed voice-band call by looking up in a routing table, which is kept in a memory in the HAS 307, a routing data block (RDB) is associated with an adjunct logical address (ALA) from the partial routing information. Then, in block 435, the HAS 307 routes the direct-dialed voice-band call to the originating IP gateway 215. Then, in block 440, the HAS 307 performs a call detail recording function and initiates the VOIP billing record, for example, creates a call detail record to track the direct-dialed voice-band call and to be added to a telephone-bill associated with the calling party's telephone number. As part of this call detail record, the HAS 307 adds a special module including the terminating access identification (ID) field, which is based on information provisioned on the trunk (circuit) groups so that the IP network use is explicitly indicated for both billing and tracking use. The HAS 307 then releases the call detail record to the standard telephone network billing system. In block 445, the direct-dialed voice-band call is received by the originating IP gateway 215. In block 450, the direct-dialed voice-band call setup is converted to packets. In block 455, the partial routing information is used by the originating IP gateway to determine the specific routing information for the destination IP gateway. In block 460, the packets are routed to the destination IP gateway over the IP network 110.

In block 465, the packets are received at the destination IP gateway, for example, IP gateway 130, and, then in block 470, the packets are reassembled in their correct order and converted to a reassembled voice-band call. In block 475, the reassembled voice-band call is routed to the local-access-provider network 340. In block 480, the local-access-provider network 340 receives the reassembled voice-band call and routes the reassembled voice-band call to the destination telephone number 136. In block 485, the reassembled voice-band call is received, connected to and conducted by the destination telephone number. Then, in block 487, the VOIP call is in progress with transmissions going back and forth between the calling party's telephone number and the destination telephone number. In block 490, the HAS 307 receives a notice of call clearing after the VOIP call is completed, which can be signaled when one or both of the calling party and called party disconnect from the VOIP call or some portion of the network drops the VOIP call. In block 495, the telephone billing system also receives notice of the call clearing and closes the call detail record.

For the VOIP service to operate each customer desiring to place VOIP calls must register/sign-up for the VOIP service and have the system store a VOIP service registration record for the calling party prior to the calling party placing a direct-dialed voice-band call. However, this sign-up process is only done once, just as with other calling plans. Similarly, another preliminary activity involves storing an allowable destination number list, which identifies numbers accessible using the VOIP service, prior to the calling party placing the direct-dialed telephone call.

While the above detailed description of the method of operation of the VOIP network system has been described in reference to the embodiment illustrated in FIG. 3, it is not intended to limit the scope of the invention as other embodiments are contemplated in accordance with the present invention.

FIG. 5 depicts a detailed functional flow diagram of the operation detailed in FIG. 4 for the direct-dialed VOIP apparatus of the entire direct-dialed VOIP network system of FIG. 3 according to an embodiment to the present invention.

In FIG. 5, in block 505 a direct-dialed voice-band call is received by the OAS 305 from the calling party's telephone number, where the direct-dialed voice-band call is associated with a destination phone number. In block 510, a test is performed using the USDS 310 to determine if the calling party's telephone number was registered for the VOIP service. If the calling party's telephone number was not registered for the VOIP service, then, in block 512, the voice-band call is routed as a circuit-switched network call and, in block 514, a standard circuit-switched billing record is initiated. If the calling party's telephone number was registered for the VOIP service, then, in block 515, a test is performed using the USDS 310 to determine if the destination number of the voice-band call is served by an IP gateway. If the destination number of the voice-band call is not served by an IP gateway, then, in block 512, the voice-band call is routed as a circuit-switched network call and, in block 514, a standard circuit-switched billing record is initiated. If the calling party's telephone number was registered for the VOIP service, then, in block 520 partial routing information is received by the OAS 305 and, in block 525, the direct-dialed voice-band call and partial routing information are routed to the HAS 307 from the OAS 305. In block 530, the direct-dialed voice-band call and partial routing information are received by the HAS 307. In block 535, the

direct-dialed voice-band call is routed to the originating IP gateway 215 by the HAS 307. In block 540, the HAS 307 initiates an IP billing record by creating a call detail record to track the direct-dialed voice-band call and be added to a telephone-bill associated with the calling party's telephone number. In block 545, the HAS 307 receives notice of the destination telephone number connecting to the direct-dialed voice-band call. Then, in block 550, the VOIP call is in progress with transmissions going back and forth between the calling party's telephone number and the destination telephone number. In block 555, the HAS 307 receives a notice of call clearing after the VOIP call is completed, which can be signaled when one or both of the calling party and called party disconnect from the VOIP call or some portion of the network drops the VOIP call. In block 560, the telephone billing system also receives notice of the call clearing and closes the call detail record.

While the above detailed description of the method of operation of the VOIP apparatus has been described in reference to the embodiment illustrated in FIG. 3, it is not intended to limit the scope of the invention as other embodiments are contemplated in accordance with the present invention.

FIG. 6 depicts a detailed block diagram of an offline provisioning system that can be configured to operate with the direct-dialed VOIP service, in accordance with an embodiment to the present invention. In FIG. 6, a network provisioning platform (NPP) 605, which is configured to receive a customer order for the VOIP service and keep the network and the billing systems synchronized, is communicatively linked to a billing system 610, which is configured to maintain customer account information and the VOIP calling plan. The NPP 605 is also communicatively linked to a Customer Service Message System (CSMS) 615, which is configured to provision customer calling number and service in the VOIP network. The CSMS 615 is also communicatively

linked to a Universal Subscriber Data Structure (USDS) database 620 and a 4ESS telecommunications switch 625. The 4ESS 625 is communicatively linked to a data record system 630, which is configured to store call detail records for completed VOIP calls with a terminating access ID equal to "IP".

Specifically, in FIG. 6, the NPP 605 is configured to receive a VOIP service registration/order for the calling party's telephone number, generate a billing system order and a network service order for the calling party's telephone number VOIP service and, then, forward the orders to the billing system 610 and the CSMS 615, respectively. The NPP 605 is also configured to manage the interaction between a calling party telephone number and a billed telephone number, to provide updated records to the billing system 610 and the CSMS 615 to compensate for numbering plan changes. The NPP 605 is further configured to synchronize any other changes due to calling party activations, disconnections and/or changes in the billing system 610 and the CSMS 615.

The billing system 610 is further configured to maintain calling party telephone number account information, to maintain a rating table with applicable PSTN and VOIP rates, and to create bills using the rating table, use records, terminating access ID and calling plan uniform service order code (USOC).

The CSMS 615 is further configured to synchronize between the 4ESS telecommunications switch 625 and the network database in the USDS 620, which stores the calling party telephone numbers that are registered for the VOIP service, USOC information and destination number information.

While the above detailed description of the offline provisioning system has been described in reference to the embodiment illustrated in FIG. 6, it is not intended to limit the scope of the invention as other embodiments are contemplated in accordance with the present invention.

FIG. 7 depicts a detailed functional flow diagram of the operation of a scalable offline provisioning system in FIG. 6 according to an embodiment to the present invention. The system is scalable because it can be configured to work with both small (for example, a corporate telephone network on a LAN) and large systems (for example, the national and international system illustrated in FIG. 3). In FIG. 7, in block 705, the NPP 605 receives a VOIP service registration/order for a calling party telephone number and, then in block 710, the NPP 605 generates a billing system order and a network service order for the calling party telephone number's VOIP service. In block 715, the NPP 605 forwards the billing system order to the billing system 610 to be stored and the network system order to the CSMS 615 to be stored. In block 720, the NPP 605 manages the billing interaction for a billed account between at least one calling party telephone number and a billed telephone number when a call is placed from the calling party telephone number. That is, the NPP 605 ensures that the call detail record generated for each call is billed to the correct telephone number after the call is terminated. In general, the calling party telephone number and the billed telephone number are the same number, however, these numbers can also be different numbers. In block 725, the NPP 605 provides updated records to the billing system 610 and the CSMS 615 to reflect numbering plan changes. In block 730, the NPP 605 synchronizes the changes between the billing system 610 and the CSMS 615 due to, for example, calling party activations, disconnections and other changes. In block 735, the billing system 610 processes the call detail records from completed

calls, where the call detail records include information on the terminating access ID, VOIP rating and bill notification.

The CSMS 615 is further configured to synchronize between the telecommunication switch 625 and the network database in the USDS 620, which stores the calling party telephone numbers that are registered for the VOIP service, network service information (for example, USOC information) and destination number information.

While the above detailed description of the method of operation of the offline provisioning system has been described in reference to the embodiment illustrated in FIG. 6, it is not intended to limit the scope of the invention as other embodiments are contemplated in accordance with the present invention. For example, embodiments of the present can be implemented to handle intra-state, inter-state and international voice-band calls (for example, regular telephone calls, facsimile transmissions and modem initiated calls) using standard circuit-switched telephone lines, cable, twisted pair, digital subscriber line (DSL) and wireless. Similarly, in an embodiment of the present invention, the customer can register for the VOIP service by selecting both a provider, that is a Primary Interexchange Carrier (PIC) and a calling plan. Alternatively, in another embodiment of the present invention, the customer can register for the VOIP service by only selecting a PIC, for example, if the PIC does not need to have a calling plan associated with the PIC in order to provide the VOIP service. The select the PIC only embodiment could be used, for example, for a provider that only provides the VOIP service.

In an embodiment of the present invention, a method for routing direct-dialed voice-band calls over an IP network includes receiving a direct-dialed voice-band call from a calling party telephone number, the direct-dialed voice-band call being associated with a destination number. The method further

includes automatically routing the direct-dialed voice-band call to the destination number as a VOIP telephone call if the calling party telephone number is registered for a VOIP service and if the destination number of the direct-dialed telephone call is accessible by the VOIP service.

In an embodiment of the present invention, a method for automatically provisioning and maintaining a network system for routing direct-dialed voice-band calls from a calling party telephone number over an IP network includes receiving a VOIP service registration for the calling party telephone number, generating at least one order record for the calling party telephone number's VOIP service and storing the at least one order record for the calling party telephone number's VOIP service. The method further includes managing the billing interaction for a billed account between at least one calling party telephone number and a billed telephone number; synchronizing changes made to the stored at least one order record for the calling party telephone number's VOIP service, between the network system and a billing system, due to calling party activations, disconnections and changes; and processing at least one call detail record including at least a terminating access identification (ID).

In an embodiment of the present invention, an apparatus includes a first voice-band switch and a database coupled to the first voice-band switch. In the apparatus, the first voice-band switch is configured to receive a direct-dialed voice-band call from a calling party's telephone number and to automatically designate the direct-dialed voice-band call as a VOIP call.

In an embodiment of the present invention, an apparatus includes an ESS OAS, with the OAS being configured to receive a direct-dialed voice-band call from a calling party's telephone number, the direct-dialed voice-band call being associated with a destination telephone number. The OAS is further

configured to determine whether to route the direct-dialed voice-band call over an IP network or a circuit-switched network and, if it is determined to route the direct-dialed voice-band call over the IP network, the OAS is configured to transmit the direct-dialed voice-band call to the IP network, or, if it is determined to continue to route the direct-dialed voice-band call over the circuit-switched network, the OAS is configured to transmit the direct-dialed voice-band call to the circuit-switched network. The apparatus further includes an USDS coupled to the ESS OAS, with the USDS being configured to store information on a plurality of calling party's telephone numbers registered for the VOIP service, to store information on which destination telephone numbers are accessible using the VOIP service, to receive the calling party's telephone number and a destination telephone number of the direct-dialed voice-band call from the OAS, to determine if the calling party's telephone number is registered for the VOIP service, and, if the calling party's telephone number is registered for the VOIP service, to determine if the destination telephone number is accessible using the VOIP service, and to return a partial routing instruction and service information to the OAS.

In an embodiment of the present invention, a system for automatically provisioning and maintaining a network system for routing direct-dialed voice-band calls from a calling party telephone number over an IP network includes a network provisioning component. The network provisioning component is configured to receive a VOIP service registration for the calling party telephone number, to generate at least one order record for the calling party telephone number's VOIP service, to store the at least one order record for the calling party telephone number's VOIP service, to manage the billing interaction for a billed account between at least one calling party telephone number and a billed telephone number, and to update the at least one order record to compensate

for numbering plan changes. The system further includes a billing system component coupled to the network provisioning component, and the billing system component is configured to maintain at least one calling party's account information, to maintain the VOIP service, to create bills based on usage, terminating access ID and calling plan USOC. The system further includes a CSMS component, which is coupled to the network provisioning component. The CSMS component is configured to synchronize changes made to the stored at least one order record for the calling party telephone number's VOIP service, between at least one telecommunications switch and a database, which stores calling party telephone numbers that are registered for the VOIP service, USOC information and destination number information. The network provisioning component is further configured to synchronize changes made to the stored at least one order record for the calling party telephone number's VOIP service, between the network system and a billing system, due to calling party activations, disconnections and changes.

In the foregoing detailed description and figures, several embodiments of the present invention are specifically illustrated and described. Accordingly, it will be appreciated that modifications and variations of the present invention are covered by the above teachings and within the purview of the appended claims without departing from the spirit and intended scope of the invention.

What Is Claimed is:

1. A method for routing direct-dialed voice-band calls over an Internet protocol (IP) network, comprising:

receiving a direct-dialed voice-band call from a calling party telephone number, the direct-dialed voice-band call being associated with a destination number; and

automatically routing the direct-dialed voice-band call to be routed to the destination number as a voice-over-Internet protocol (VOIP) telephone call if the calling party telephone number is registered for a VOIP service and if the destination number of the direct-dialed telephone call is accessible by the VOIP service.

2. The method of claim 1 further comprising:

receiving a registration for the calling party's telephone number for the VOIP service prior to the calling party placing the direct-dialed telephone call.

3. The method of claim 2 further comprising:

storing a VOIP service registration record for the calling party telephone number.

4. The method of claim 2 further comprising:

storing an allowable destination list, which identifies the destination numbers accessible using the VOIP service, prior to the calling party placing the direct-dialed telephone call.

5. The method of claim 1, wherein automatically routing the direct-dialed voice-band call to the destination number as a voice-over-Internet protocol (VOIP) telephone call if the calling party telephone number is registered for a VOIP service and the destination number of the direct-dialed telephone call is accessible by the VOIP service comprises:

determining if a VOIP service registration record for the calling party telephone number exists; and

if the VOIP service registration record for the calling party telephone number exists, determining if the destination number of the direct-dialed voice-band call is accessible by the VOIP service.

6. The method of claim 5 further comprising:

if the calling party's telephone number is registered for the VOIP service and if the destination number of the direct-dialed voice-band call is accessible by the VOIP service, receiving an indication to route the direct-dialed voice-band call over the IP network, otherwise, receiving an indication to route the direct-dialed voice-band call over a circuit-switched public switched telephone network (PSTN).

7. The method of claim 6 further comprising:

initiating a billing record for the direct-dialed voice-band call if the indication is to route the direct-dialed voice-band call over the IP network, wherein the billing record is associated with the calling party's telephone number.

8. The method of claim 7 further comprising:

routing the direct-dialed voice-band call to the IP network

9. The method of claim 8 further comprising:

receiving notice of the direct-dialed voice-band call clearing.

10. The method of claim 9 further comprising:

closing the billing record for the direct-dialed voice-band call.

11. An article of manufacture comprising a computer-readable medium having stored thereon instructions adapted to be executed by a processor, the instructions which, when executed, define a series of steps to route direct-dialed voice-band calls over an Internet protocol (IP) network, said steps comprising:

receiving a direct-dialed voice-band call from a calling party telephone number, the direct-dialed voice-band call being associated with a destination number; and

automatically routing the direct-dialed voice-band call to the destination number as a voice-over-Internet protocol (VOIP) telephone call if the calling party telephone number is registered for a VOIP service and if the destination number of the direct-dialed telephone call is accessible by the VOIP service.

12. The computer-readable medium of claim 11, wherein said steps further comprise:

receiving a registration for the calling party's telephone number for the VOIP service prior to the calling party placing the direct-dialed telephone call.

13. The computer-readable medium of claim 12, wherein said steps further comprise:

storing a VOIP service registration record for the calling party telephone number.

14. The computer-readable medium of claim 12, wherein said steps further comprise:

storing an allowable destination list, which identifies the destination numbers accessible using the VOIP service, prior to the calling party placing the direct-dialed telephone call.

15. A method for automatically provisioning and maintaining a network system for routing direct-dialed voice-band calls from a calling party telephone number over an Internet protocol (IP) network comprising:

receiving a voice-over-Internet protocol (VOIP) service registration for the calling party telephone number;

generating at least one order record for the calling party telephone number's VOIP service;

storing the at least one order record for the calling party telephone number's VOIP service;

managing the billing interaction for a billed account between at least one calling party telephone number and a billed telephone number;

synchronizing changes made to the stored at least one order record for the calling party telephone number's VOIP service, between the network system and a billing system, due to calling party activations, disconnections and changes; and

processing at least one call detail record including at least a terminating access identification (ID).

16. The method of claim 15 further comprising:

updating the at least one order record to compensate for numbering plan changes.

17. The method of claim 15, wherein generating each of the at least one order records for the calling party telephone number's VOIP service comprises:

generating a billing system order record; and

generating a network order record.

18. The method of claim 15, wherein storing each of the at least one order records for the calling party telephone number's VOIP service comprises:

storing the billing system order record; and

storing the network order record.

19. The method of claim 17, wherein updating each of the at least one order records to compensate for numbering plan changes comprises:

updating the billing system order record; and

updating the network order record.

20. The method of claim 15, wherein synchronizing changes due to calling party activations, disconnections and changes between the network system and a billing system comprises:

ensuring for each calling party telephone number registered for the VOIP service that the billing system order record and network order record both reflect the same numbering plan changes, activations, disconnections and other changes.

21. The method of claim 15 further comprising:

receiving a direct-dialed voice-band call to a destination number from a calling party telephone number, the direct-dialed voice-band call being associated with a destination number; and

automatically routing the direct-dialed voice-band call to the destination number as a voice-over-Internet protocol (VOIP) telephone call if the calling party telephone number is registered for a VOIP service and if the destination number of the direct-dialed telephone call is accessible by the VOIP service.

22. The method of claim 21 further comprising:

receiving a registration for the calling party's telephone number for the VOIP service prior to the calling party placing the direct-dialed telephone call.

23. The method of claim 22 further comprising:

storing a VOIP service registration record for the calling party telephone number.

24. The method of claim 21 further comprising:

storing an allowable destination number list, which identifies numbers accessible using the VOIP service, prior to the calling party placing the direct-dialed telephone call.

25. The method of claim 21, wherein automatically routing the direct-dialed voice-band call to the destination number as a voice-over-Internet protocol (VOIP) telephone call if the calling party telephone number is registered for a VOIP service and the destination number of the direct-dialed telephone call is accessible by the VOIP service comprises:

determining if a VOIP service registration record for the calling party telephone number exists; and

if the VOIP service registration record for the calling party telephone number exists, determining if the destination number of the direct-dialed voice-band call is accessible by the VOIP service.

26. The method of claim 25 further comprising:

if the calling party's telephone number is registered for the VOIP service and if the destination number of the direct-dialed voice-band call is accessible by the VOIP service, receiving an indication to route the direct-dialed voice-band call over the IP network, otherwise, receiving an indication to route the direct-dialed voice-band call over a circuit-switched public switched telephone network (PSTN).

27. The method of claim 26 further comprising:

initiating a billing record for the direct-dialed voice-band call if the indication is to route the direct-dialed voice-band call over the IP network, wherein the billing record is associated with the calling party's telephone number.

28. The method of claim 27 further comprising:

routing the direct-dialed voice-band call to the IP network

29. The method of claim 28 further comprising:

receiving notice of the direct-dialed voice-band call clearing.

30. The method of claim 29 further comprising:

closing the billing record for the direct-dialed voice-band call.

31. An article of manufacture comprising a computer-readable medium having stored thereon instructions adapted to be executed by a processor, the instructions which, when executed, define a series of steps to automatically provision and maintain a network system for routing direct-dialed voice-band calls from a calling party telephone number over an Internet protocol (IP) network, said steps comprising:

receiving a voice-over-Internet protocol (VOIP) service registration for the calling party telephone number;

generating at least one order record for the calling party telephone number's VOIP service;

storing the at least one order record for the calling party telephone number's VOIP service;

managing the billing interaction for a billed account between at least one calling party telephone number and a billed telephone number;

synchronizing changes made to the stored at least one order record for the calling party telephone number's VOIP service, between the network system and a billing system, due to calling party activations, disconnections and changes; and

processing at least one call detail record including at least a terminating access identification (ID).

32. The article of manufacture of claim 31 further comprising:

updating the at least one order record to compensate for numbering plan changes.

33. The article of manufacture of claim 31, wherein said generating each of the at least one order records for the calling party telephone number's VOIP service step further comprises:

generating a billing system order record; and

generating a network order record.

34. The article of manufacture of claim 33, wherein said storing each of the at least one order records for the calling party telephone number's VOIP service step further comprises:

storing the billing system order record; and

storing the network order record.

35. The article of manufacture of claim 34, wherein updating each of the at least one order records to compensate for numbering plan changes step further comprises:

updating the billing system order record; and

updating the network order record.

36. The article of manufacture of claim 34, wherein synchronizing changes due to calling party activations, disconnections and changes between the network system and a billing system step further comprises:

ensuring for each calling party telephone number registered for the VOIP service that the billing system order record and network order record both reflect the same numbering plan changes, activations, disconnections and other changes.

37. An apparatus, comprising:

a first voice-band switch; and

a database coupled to the first voice-band switch;

the first voice-band switch being configured to receive a direct-dialed voice-band call from a calling party's telephone number and to automatically designate the direct-dialed voice-band call as a voice-over-Internet protocol (VOIP) call.

38. The apparatus of claim 37, wherein the first voice-band switch is further configured to automatically route the direct-dialed voice-band call using the VOIP service when the direct-dialed voice-band call is designated as a VOIP call.

39. The apparatus of claim 38, wherein the first voice-band switch is an electronic switching system (ESS) originating assist switch (OAS), the database is a universal subscriber data structure (USDS) and the first voice-band switch is communicatively linked to an IP gateway.

40. The apparatus of claim 38, wherein the first voice-band switch is further configured to open a billing record for the VOIP call.

41. The apparatus of claim 37, wherein the first voice-band switch is configured to automatically designate the direct-dialed voice-band call as the VOIP call if the database contains information that the calling party's telephone number is registered for a VOIP service and if a destination number of the direct-dialed voice-band call is accessible by the VOIP service.

42. The apparatus of claim 37, wherein the first voice-band switch is further configured to route the VOIP call to an IP gateway for routing to an IP network, then to a destination IP gateway and then to a local access provider network for routing to a destination number of the direct-dialed voice-band call.

43. The apparatus of claim 37, wherein the first voice-band switch is further configured, if the database contains information that the calling party's telephone number is only registered for non-single-stage VOIP services or if the database contains information that the calling party's telephone number is registered for the single-stage VOIP service and the destination number of the direct-dialed voice-band call is inaccessible by the VOIP service, to automatically designate the direct-dialed voice-band call as a circuit-switched call; and to automatically route the direct-dialed voice-band call for routing as a

circuit-switched call if the direct-dialed voice-band call is designated as a circuit-switched call.

44. The apparatus of claim 37, further comprising:

a provisioning system configured to automatically provision and maintain the network apparatus.

45. The apparatus of claim 44, wherein the provisioning system comprises:

a network provisioning platform (NPP) configured to receive a voice-over-Internet protocol (VOIP) service registration for the calling party, to generate at least one order for the calling party's VOIP service, to store the at least one order for the calling party telephone number's VOIP service, to manage the interaction between a calling party's telephone number and a billed telephone number, and to update a plurality of service records to compensate for numbering plan changes;

a billing system coupled to the NPP, wherein the billing system is configured to maintain at least one calling party's account information, to maintain the VOIP service, and to create bills based on usage, a terminating access ID and a calling plan uniform service order code (USOC); and

a customer service message system (CSMS) coupled to the NPP, wherein the CSMS is configured to synchronize between the first voice-band switch and the database which is configured to store calling party telephone numbers that are registered for the VOIP service, USOC information and destination number information;

wherein the NPP is further configured to synchronize changes in the network system and the billing system due to calling party activations, disconnections and changes.

46. The apparatus of claim 45, wherein the CSMS is further configured to administer in the database at least one of the group including a country code field, a destination telephone number field, and a destination code field.

47. The apparatus of claim 38 further comprising:

a second voice-band switch coupled to the first voice-band switch, wherein the second voice-band switch is configured to receive the automatically routed direct-dialed voice-band call from the first voice-band switch, to forward the direct-dialed voice-band call for transmission as a VOIP call, and to open a billing record for the VOIP call.

48. The apparatus of claim 47, wherein the first voice-band switch is further configured, if the database contains information that the calling party's telephone number is only registered for non-single-stage VOIP services or if the database contains information that the calling party's telephone number is registered for the single-stage VOIP service and the destination number of the direct-dialed voice-band call is inaccessible by the VOIP service, to automatically designate the direct-dialed voice-band call as a circuit-switched call; and to automatically route the direct-dialed voice-band call for routing as a circuit-switched call if the direct-dialed voice-band call is designated as a circuit-switched call.

49. The apparatus of claim 47, wherein the second voice-band switch is further configured to route the direct-dialed call to an IP gateway for routing to an IP network, then to a destination IP gateway and then to a local access provider network for routing to a destination number of the direct-dialed voice-band call.

50. The apparatus of claim 47, wherein the first voice-band switch is an electronic switching system (ESS) originating assist switch (OAS), the second voice-band switch is an ESS handoff assist switch (HAS), the database is a universal subscriber data structure (USDS) and the second voice-band switch is communicatively linked to an IP gateway.

51. The apparatus of claim 48, wherein the first voice-band switch is further configured to automatically route the direct-dialed voice-band call as a circuit-switched call if the direct-dialed voice-band call is to be routed as a circuit-switched call.

52. The apparatus of claim 48, further comprising:

a provisioning system configured to automatically provision and maintain the network apparatus.

53. The apparatus of claim 52, wherein the provisioning system comprises:

a network provisioning platform (NPP) configured to receive a voice-over-Internet protocol (VOIP) service registration for the calling party, to generate at least one order for the calling party's VOIP service, to store the at least one order for the calling party's VOIP service, to manage the interaction

between a billed telephone number and a calling party's telephone number, and to update a plurality of calling party records to compensate for numbering plan changes;

a billing system coupled to the NPP, wherein the billing system is configured to maintain at least one calling party's account information, to maintain the VOIP service, and to create bills based on usage, a terminating access ID and a calling plan uniform service order code (USOC); and

a customer service message system (CSMS) coupled to the NPP, wherein the CSMS is configured to synchronize between the first voice-band switch and the database which is configured to store calling party telephone numbers that are registered for the VOIP service, USOC information and destination number information;

wherein the NPP is further configured to synchronize changes in the network system and the billing system due to calling party activations, disconnections and changes.

54. The apparatus of claim 53, wherein the CSMS is further configured to administer in the database at least one of the group including a country code field, a destination telephone number field, and a destination code field.

55. An apparatus, comprising:

an electronic switching system (ESS) originating assist switch (OAS), the OAS being configured to receive a direct-dialed voice-band call from a calling party's telephone number, the direct-dialed voice-band call being associated with a destination telephone number, to determine whether to route the direct-dialed voice-band call over an Internet protocol (IP) network or a circuit-switched network, and, if it is determined to route the direct-dialed voice-band

call over the IP network, the OAS is configured to transmit the direct-dialed voice-band call to the IP network, or, if it is determined to continue to route the direct-dialed voice-band call over the circuit-switched network, the OAS is configured to transmit the direct-dialed voice-band call to the circuit-switched network; and

an universal subscriber data structure (USDS) coupled to the ESS OAS, the USDS being configured to store service information on a plurality of calling party telephone numbers, to store information on which destination telephone numbers are accessible using a voice-over-Internet protocol (VOIP) service, to receive the calling party's telephone number and the destination telephone number of the direct-dialed voice-band call from the OAS, to determine if the calling party's telephone number is registered for the VOIP service, and, if the calling party's telephone number is registered for the VOIP service, to determine if the destination telephone number is accessible using the VOIP service, and to return a partial routing instruction and service information to the OAS.

56. The apparatus of claim 55 further comprising:

an ESS handoff assist switch (HAS) coupled to the OAS, wherein the HAS is configured to receive the direct-dialed voice-band call and to route the direct-dialed voice-band call to the VOIP network if the calling party's telephone number is registered for the VOIP service.

57. The apparatus of claim 55 further comprising:

a provisioning system configured to automatically provision and maintain the network apparatus.

58. The apparatus of claim 55 further comprising:

a network provisioning platform (NPP) configured to receive a voice-over-Internet protocol (VOIP) service registration for the calling party, to generate at least one order for the calling party's VOIP service, to store the at least one order for the calling party's VOIP service, to manage the billing interaction for a billed account between at least one calling party telephone number and a billed telephone number, to update a plurality of calling party records to compensate for numbering plan changes, and to synchronize changes made to the stored at least one order record for the calling party telephone number's VOIP service, between the network system and a billing system, due to calling party activations, disconnections and changes;

a billing system coupled to the NPP, wherein the billing system is configured to maintain at least one calling party's account information, to maintain the VOIP service, and to create bills based on usage, a terminating access ID and a calling plan uniform service order code (USOC); and

a customer service message system (CSMS) coupled to the NPP, wherein the CSMS is configured to synchronize between the first voice-band switch and a database configured to store calling party telephone numbers, USOC information and destination number information.

59. The apparatus of claim 55, wherein the CSMS is further configured to administer a country code field which is stored in the database.

60. A system for automatically provisioning and maintaining a network system for routing direct-dialed voice-band calls from a calling party telephone number over an Internet protocol (IP) network comprising:

a network provisioning component configured to receive a voice-over-Internet protocol (VOIP) service registration for the calling party telephone number, to generate at least one order record for the calling party telephone number's VOIP service, to store the at least one order record for the calling party telephone number's VOIP service, to manage the billing interaction for a billed account between at least one calling party telephone number and a billed telephone number, and to update the at least one order record to compensate for numbering plan changes;

a billing system component coupled to the network provisioning component, wherein the billing system component is configured to maintain at least one calling party's account information, to maintain the VOIP service, to create bills based on usage, terminating access ID and calling plan uniform service order code (USOC); and

a customer service message system (CSMS) component coupled to the network provisioning component, wherein the CSMS component is configured to synchronize changes made to the stored at least one order record for the calling party telephone number's VOIP service, between at least one telecommunications switch and a database, which stores calling party telephone numbers that are registered for the VOIP service, USOC information and destination number information;

wherein the network provisioning component is further configured to synchronize changes made to the stored at least one order record for the calling party telephone number's VOIP service, between the network system

and a billing system, due to calling party activations, disconnections and changes.

61. The system of claim 60, wherein the CSMS component is further configured to administer a country code field which is stored in the database.

62. The system of claim 60 further comprising:

a first voice-band switch; and

a database coupled to the first voice-band switch;

the first voice-band switch being configured to receive a direct-dialed voice-band call from a calling party's telephone number and to automatically designate the direct-dialed voice-band call as a voice-over-Internet protocol (VOIP) call.

63. The system of claim 62, wherein the first voice-band switch is further configured to automatically route the direct-dialed voice-band call using the VOIP service when the direct-dialed voice-band call is designated as a VOIP call.

64. The system of claim 63, wherein the first voice-band switch is an electronic switching system (ESS) originating assist switch (OAS), the database is a universal subscriber data structure (USDS) and the first voice-band switch is communicatively linked to an IP gateway.

65. The system of claim 62, wherein the first voice-band switch is further configured to open a billing record for the VOIP call.

66. The system of claim 64, wherein the first voice-band switch is further configured to automatically route the direct-dialed voice-band call as a circuit switched call if the direct-dialed voice-band call is to be routed as a circuit switched call.

67. The system of claim 64 further comprising:

a second voice-band switch coupled to the first voice-band switch, wherein, if the direct-dialed voice-band call is being routed as a VOIP call, the second voice-band switch is configured to receive the direct-dialed voice-band call from the first voice-band switch, open a billing record for the VOIP call, and to forward the direct-dialed voice-band call for transmission as a VOIP call.

68. The system of claim 64, further comprising:

a provisioning system configured to automatically provision and maintain the network apparatus.

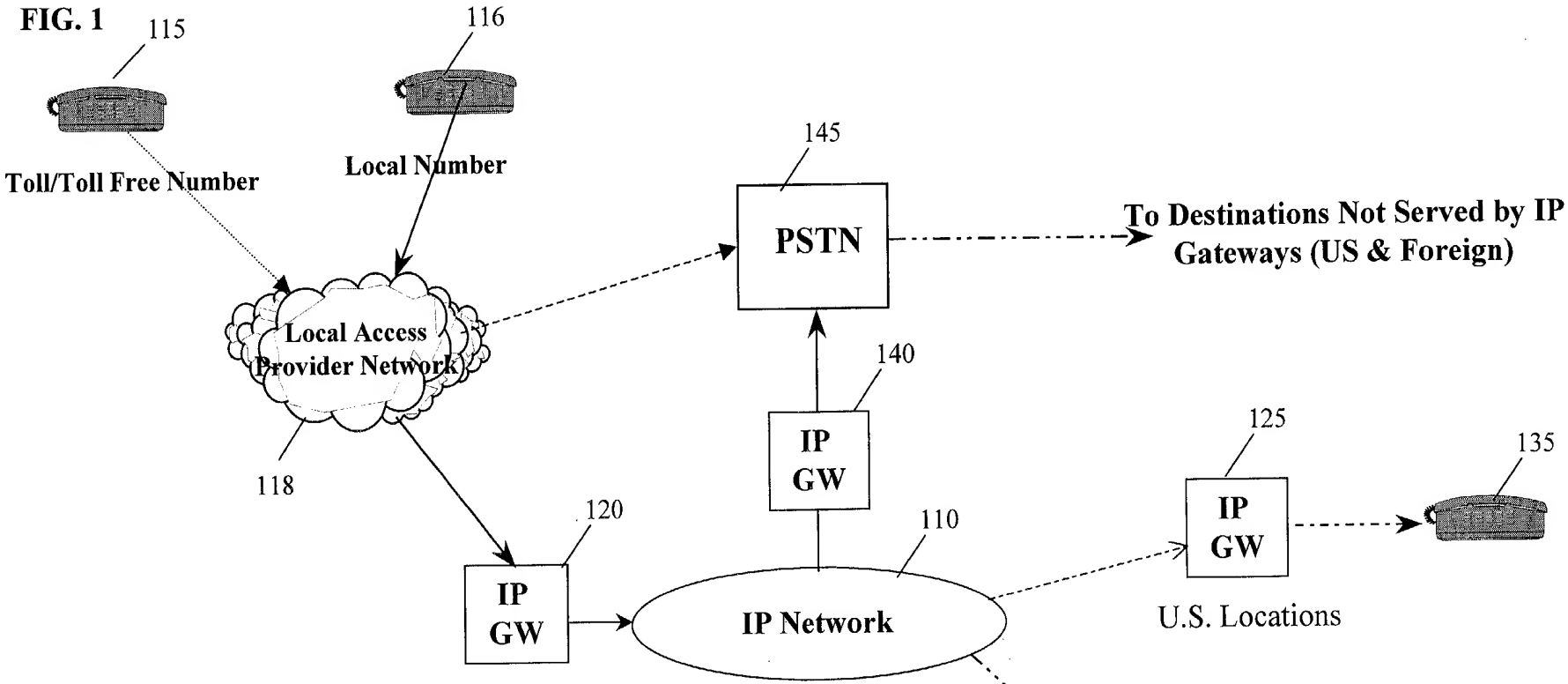
69. The system of claim 65, wherein the first voice-band switch is an electronic switching system (ESS) originating assist switch (OAS), the second voice-band switch is an ESS handoff assist switch (HAS) and the database is a universal subscriber data structure (USDS).

70. The system of claim 65, wherein the first voice-band switch is further configured to automatically route the direct-dialed voice-band call as a circuit switched call if the direct-dialed voice-band call is to be routed as a circuit switched call.

ABSTRACT OF THE DISCLOSURE

A Voice-over-Internet protocol (VOIP) communications network system that enables direct-dialed (single-stage) access to the Internet Protocol (IP) network from the circuit-switched network. Specifically, the VOIP network system includes a VOIP service implemented on a communications system which, after a customer number has been registered for the service, automatically recognizes calls from the registered customer's telephone number and determines if the call can be routed as a VOIP call over the IP network. In embodiments of the present invention, the customer can register for the VOIP service by selecting both a provider (PIC) and a calling plan or by only selecting a provider. The system can be implemented to handle intra-state, inter-state and international voice-band calls (for example, regular telephone calls, facsimile transmissions and modem initiated calls) using standard circuit-switched telephone lines, cable, twisted pair, digital subscriber line (DSL) and wireless.

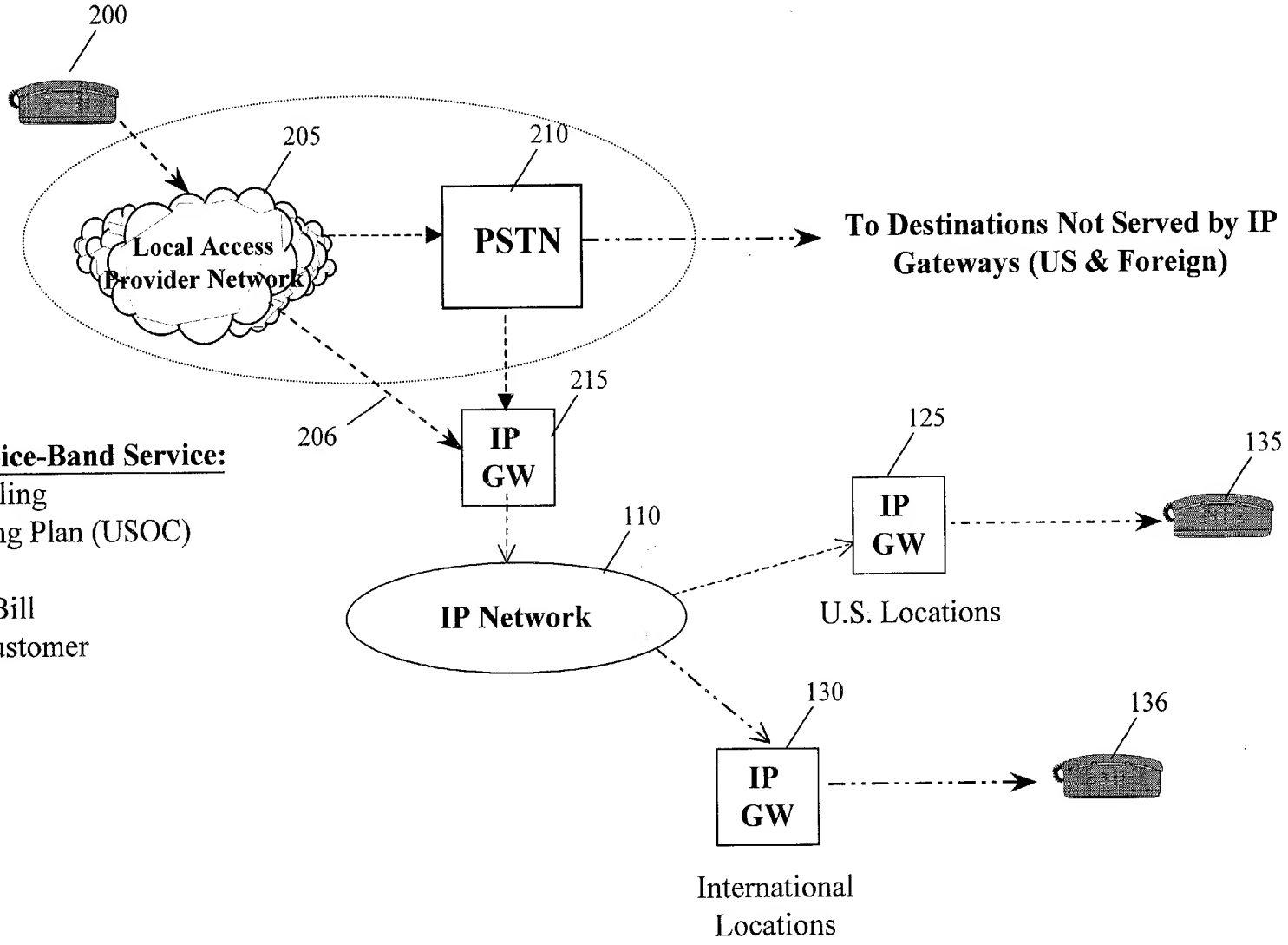
FIG. 1



Current VOIP Service:

- Local Number Access
- 2-Stage and 3-Stage Dialing Required
- Prepaid (Credit Card, Cash, Check, etc.)
- Toll/Toll Free Number Access
- Calling Party Telephone Number Verification Option
- National and International Locations

FIG. 2



Direct-Dialed Voice-Band Service:

- Single-Stage Dialing
- PIC and/or Calling Plan (USOC) Driven
- Billed to Phone Bill
- Transparent to customer

FIG. 3

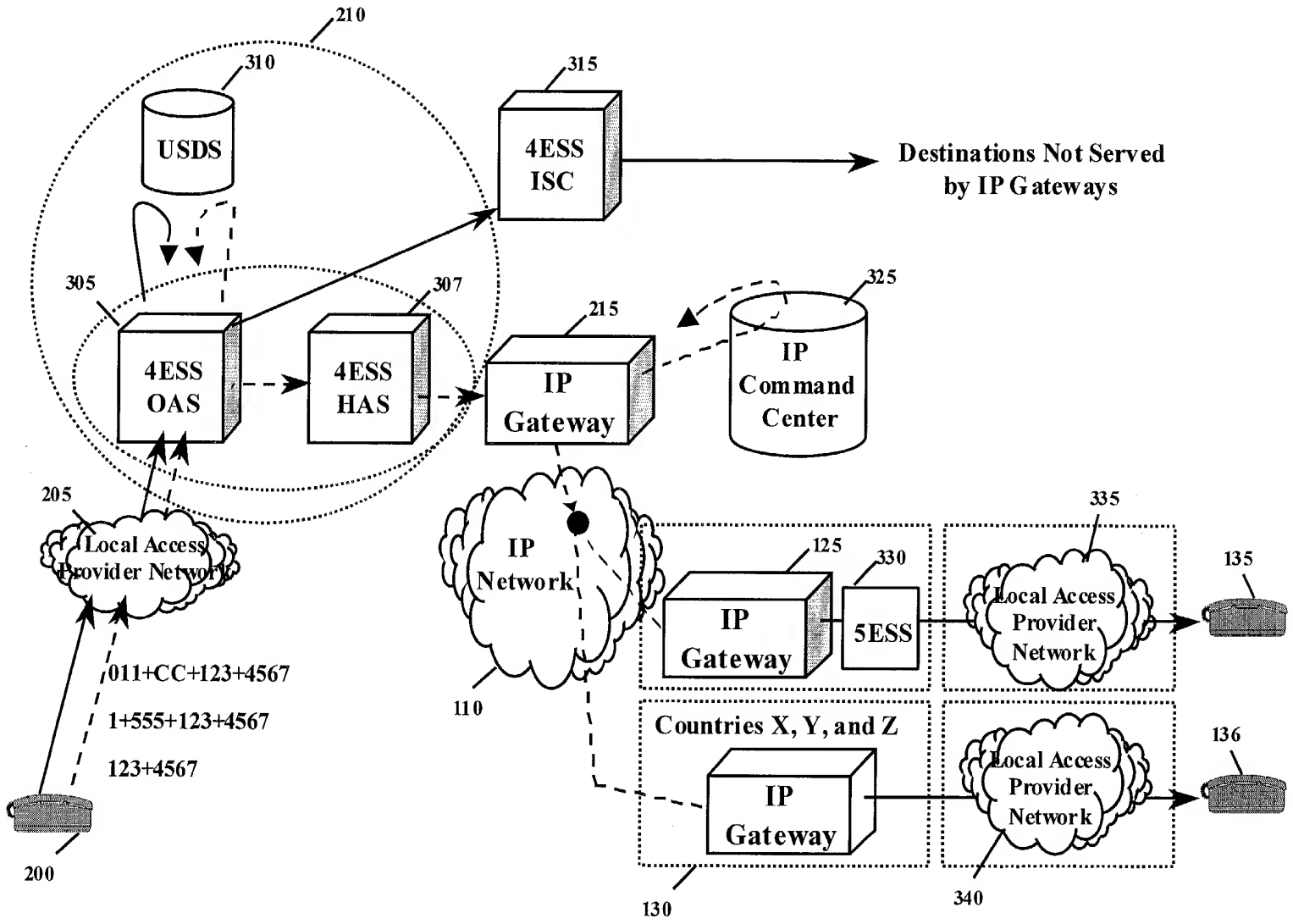

210
310
USDS
315
4ESS ISC
Destinations Not Served by IP Gateways
305
307
325
4ESS OAS
4ESS HAS
215
IP Gateway
IP Command Center
205
Local Access Provider Network
IP Network
335
125
330
IP Gateway
5ESS
Local Access Provider Network
135
011+CC+123+4567
1+555+123+4567
123+4567
110
Countries X, Y, and Z
136
IP Gateway
Local Access Provider Network
200
130
340

FIG. 4

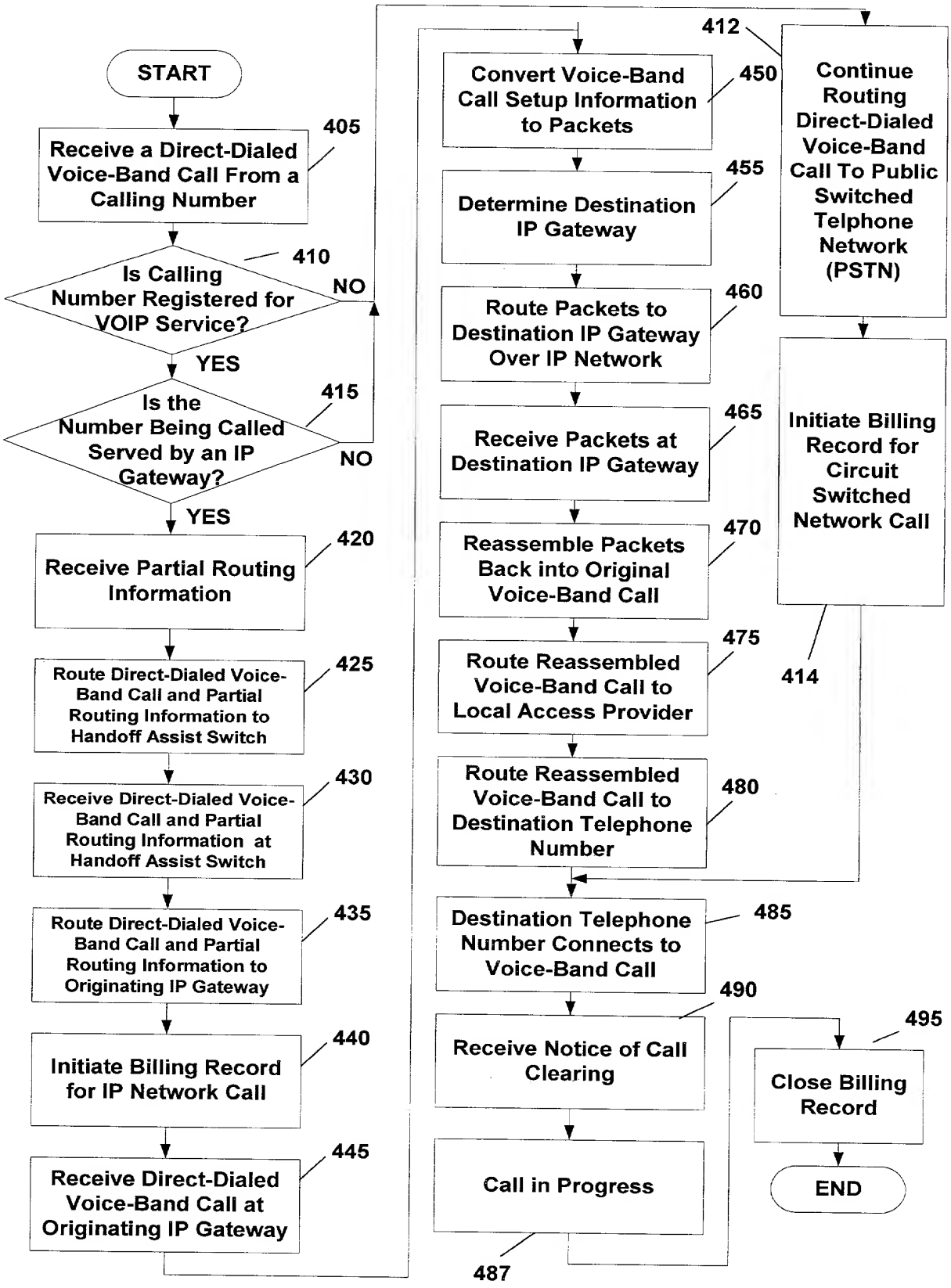

START
Receive a Direct-Dialed Voice-Band Call From a Calling Number
405
Is Calling Number Registered for VOIP Service?
410
NO
YES
Is the Number Being Called Served by an IP Gateway?
415
NO
YES
Receive Partial Routing Information
420
Route Direct-Dialed Voice-Band Call and Partial Routing Information to Handoff Assist Switch
425
Receive Direct-Dialed Voice-Band Call and Partial Routing Information at Handoff Assist Switch
430
Route Direct-Dialed Voice-Band Call and Partial Routing Information to Originating IP Gateway
435
Initiate Billing Record for IP Network Call
440
Receive Direct-Dialed Voice-Band Call at Originating IP Gateway
445
Convert Voice-Band Call Setup Information to Packets
450
Determine Destination IP Gateway
455
Route Packets to Destination IP Gateway Over IP Network
460
Receive Packets at Destination IP Gateway
465
Reassemble Packets Back into Original Voice-Band Call
470
Route Reassembled Voice-Band Call to Local Access Provider
475
Route Reassembled Voice-Band Call to Destination Telephone Number
480
Destination Telephone Number Connects to Voice-Band Call
485
Receive Notice of Call Clearing
490
Call in Progress
487
412
Continue Routing Direct-Dialed Voice-Band Call To Public Switched Telphone Network (PSTN)
Initiate Billing Record for Circuit Switched Network Call
414
495
Close Billing Record
END

FIG. 5

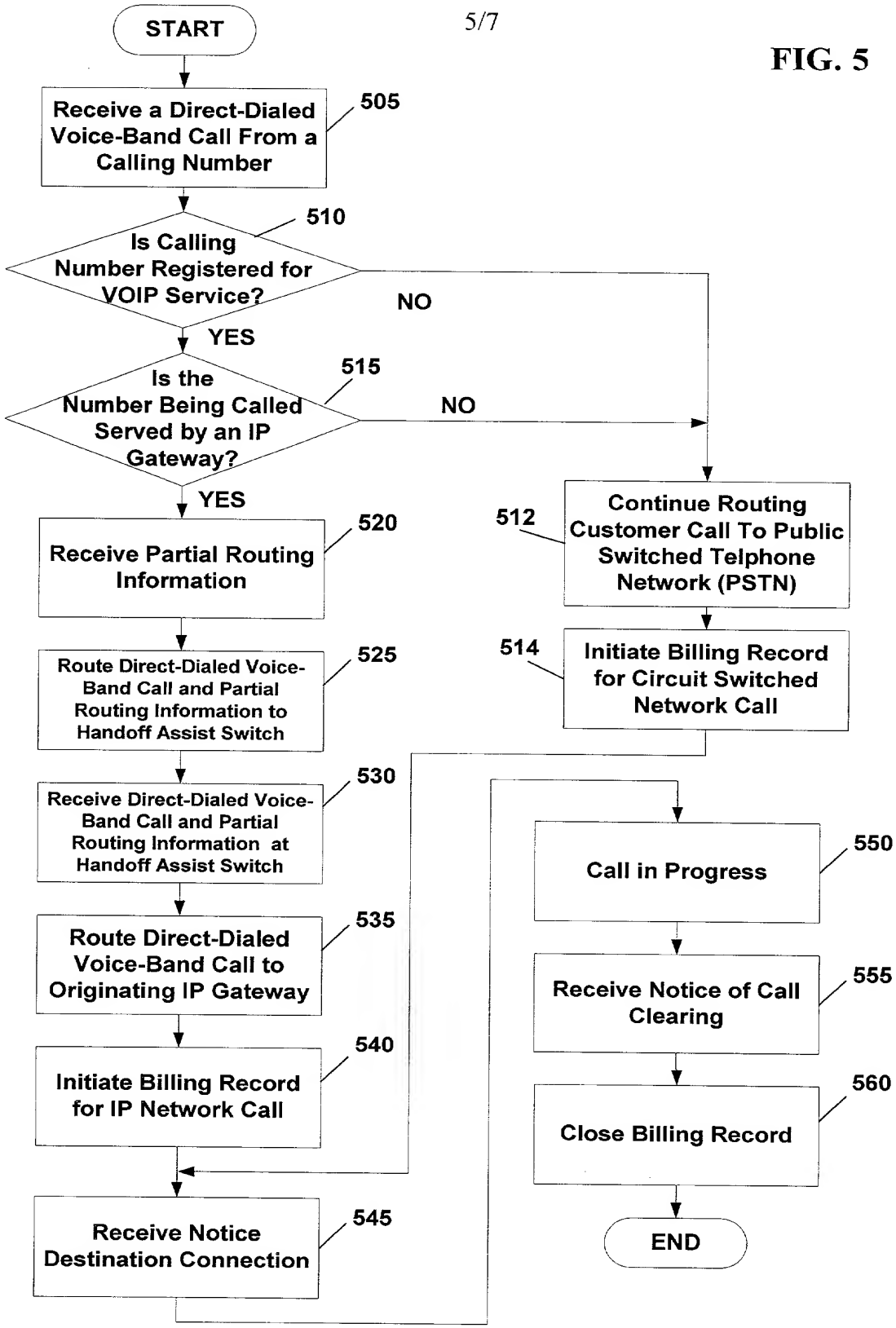

START
Receive a Direct-Dialed Voice-Band Call From a Calling Number
505
Is Calling Number Registered for VOIP Service?
510
NO
YES
Is the Number Being Called Served by an IP Gateway?
515
NO
YES
Receive Partial Routing Information
520
Route Direct-Dialed Voice-Band Call and Partial Routing Information to Handoff Assist Switch
525
Receive Direct-Dialed Voice-Band Call and Partial Routing Information at Handoff Assist Switch
530
Route Direct-Dialed Voice-Band Call to Originating IP Gateway
535
Initiate Billing Record for IP Network Call
540
Receive Notice Destination Connection
545
Continue Routing Customer Call To Public Switched Telphone Network (PSTN)
512
Initiate Billing Record for Circuit Switched Network Call
514
Call in Progress
550
Receive Notice of Call Clearing
555
Close Billing Record
560
END

FIG. 6

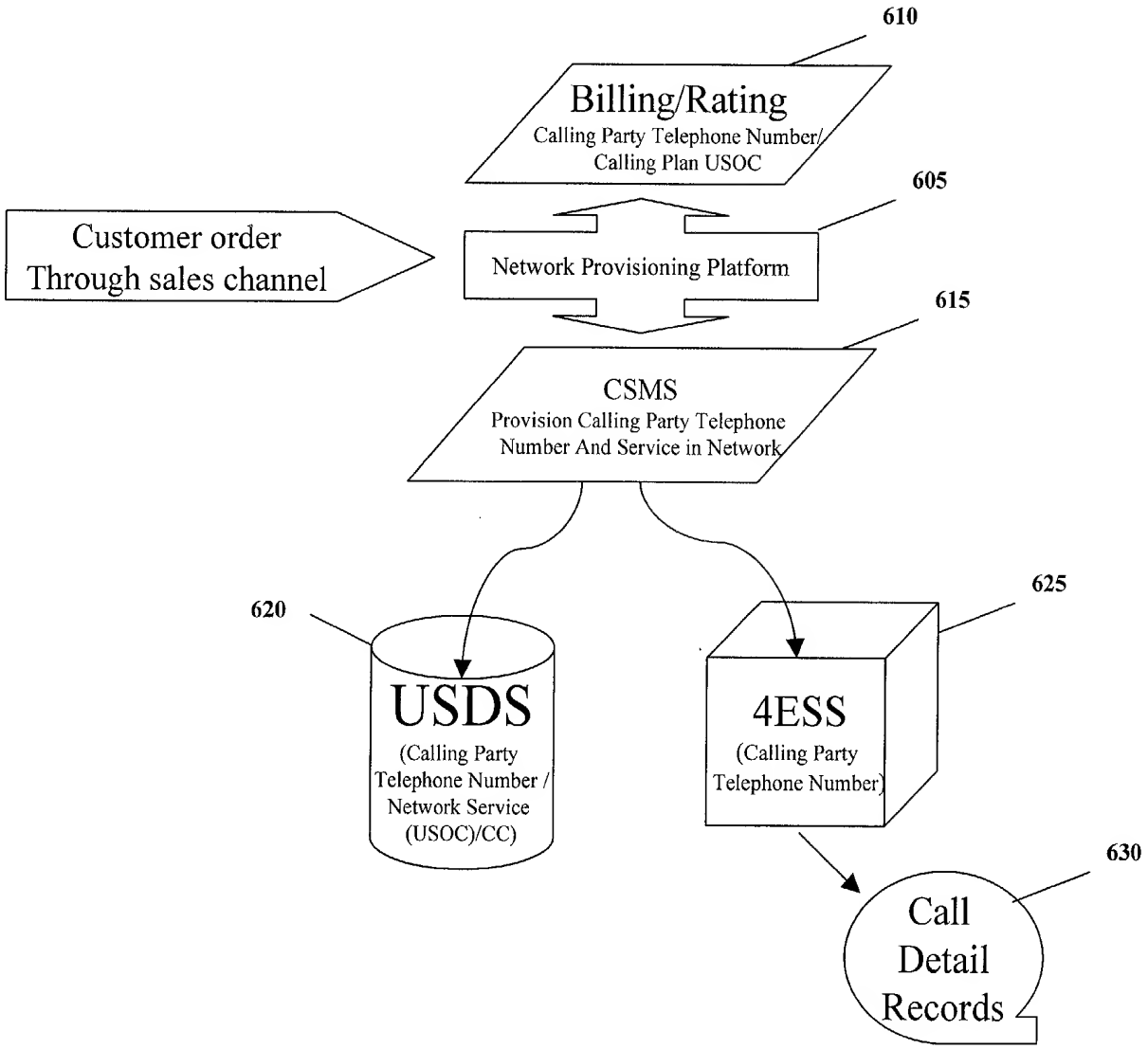
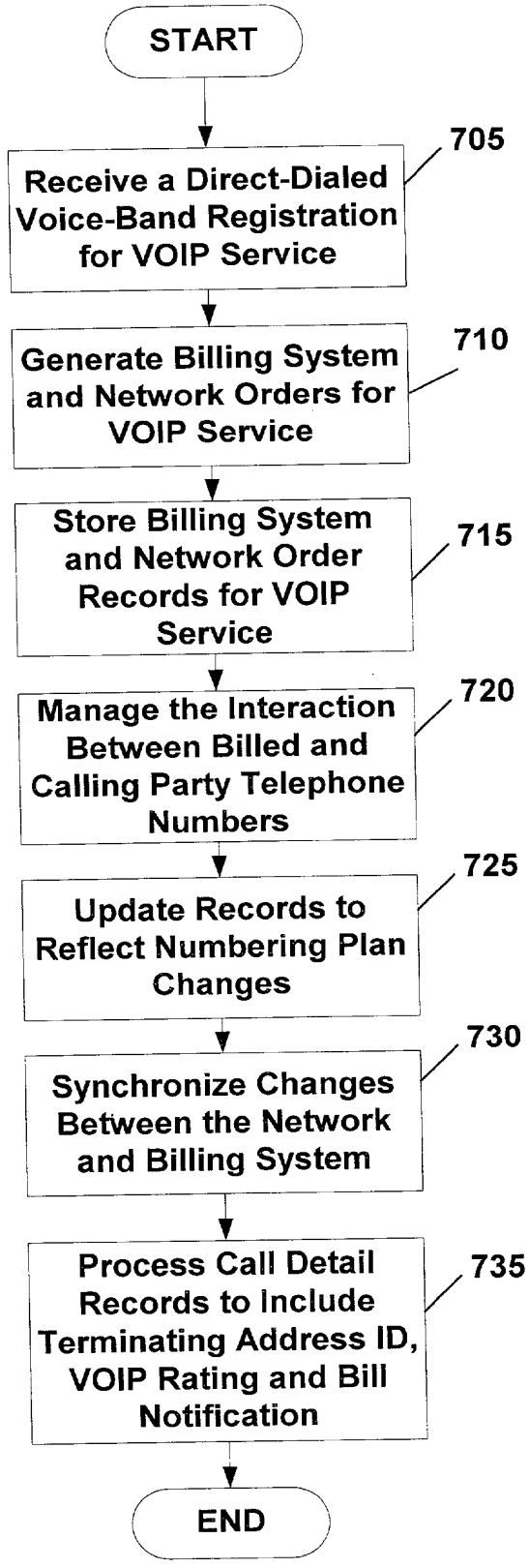

610
Billing/Rating
Calling Party Telephone Number/
Calling Plan USOC
605
Customer order
Through sales channel
Network Provisioning Platform
615
CSMS
Provision Calling Party Telephone
Number And Service in Network
620
USDS
(Calling Party
Telephone Number /
Network Service
(USOC)/CC)
625
4ESS
(Calling Party
Telephone Number)
630
Call
Detail
Records

FIG. 7


START
Receive a Direct-Dialed Voice-Band Registration for VOIP Service
705
Generate Billing System and Network Orders for VOIP Service
710
Store Billing System and Network Order Records for VOIP Service
715
Manage the Interaction Between Billed and Calling Party Telephone Numbers
720
Update Records to Reflect Numbering Plan Changes
725
Synchronize Changes Between the Network and Billing System
730
Process Call Detail Records to Include Terminating Address ID, VOIP Rating and Bill Notification
735
END

IDS No.: 1999-0566
Docket No.: 2685/5439

IN THE UNITED STATES
PATENT AND TRADEMARK OFFICE

DECLARATION AND POWER OF ATTORNEY

As below named inventors, we hereby declare that:

Our residences, post office addresses and citizenships are as stated below next to each of our names.

We believe we are first and joint inventors of the subject matter which is claimed and for which a patent is sought on the invention entitled **METHOD AND SYSTEM FOR CUSTOMER SELECTED DIRECT DIALED VOICE-OVER-INTERNET PROTOCOL (VOIP)** the specification of which is attached hereto.

We hereby state that we have reviewed and understand the contents of the above identified specification, including the claims, as amended by an amendment, if any, specifically referred to in this oath or declaration.

We acknowledge the duty to disclose all information known to us which is material to patentability as defined in Title 37, Code of Federal Regulations, 1.56.

We hereby claim foreign priority benefits under Title 35, United States Code, 119 of any foreign application(s) for patent or inventors' certificate listed below and have also identified below any foreign application for patent or inventors' certificate having a filing date before that of the application on which priority is claimed:

None

We hereby claim the benefit under Title 35, United States Code, 120 of any United States application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, 112, we acknowledge the duty to disclose all information known to us to be material to patentability as defined in Title 37, Code of Federal Regulations, 1.56 which became available between the filing date of the prior application and the national or PCT international filing date of this application:

None

We hereby declare that all statements made herein of our own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States

IDS No.: 1999-0566
Docket No.: 2685/5439

Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

We hereby appoint the following attorney(s) with full power of substitution and revocation, to prosecute said application, to make alterations and amendments therein, to receive the patent, and to transact all business in the Patent and Trademark Office connected therewith:

Samuel H. Dworetsky	(Reg. No. 27873)
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Cedric G. DeLaCruz	(Reg. No. 36498)
Michele L. Conover	(Reg. No. 34962)
Rohini K. Garg	(Reg. No. 45272)
Benjamin S. Lee	(Reg. No. 42787)
Robert B. Levy	(Reg. No. 28234)
Susan E. McHale	(Reg. No. 35948)
Alfred G. Steinmetz	(Reg. No. 22971)

We also appoint Brian S. Mudge (Reg. No. 40,738), Christopher R. Hutter (Reg. No. 41,087), Robert L. Hails, Jr. (Reg. No. 39,702) and David R. Schaffer (Reg. No. 43,089) of Kenyon & Kenyon as associate attorneys, with full power to prosecute said application, to make alterations and amendments therein, and to transact all business in the U.S. Patent and Trademark Office connected therewith.

Please address all correspondence to Mr. Samuel H. Dworetsky, AT&T Corp., 180 Park Avenue, Building 103, Room 174, Florham Park, New Jersey 07932. Telephone calls should be made to Michele L. Conover by dialing (908) 221-5773.

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IDS No.: 1999-0566
Docket No.: 2685/5439

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PATENT - 09/599,175